AMENDMENTS TO THE CLAIMS

1-12. (Canceled)

13. (Previously Presented) The method of claim 26, wherein use is made of an oscillator model for extracting signal segments from the first signal frame, the oscillator model including a codebook in which vectors of samples forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

14. (Previously Presented) The method of claim 13, wherein the second-listed producing step comprises a step of matching a true state of a trailing part of the first signal frame with said states in said codebook, and reading out a signal segment from said codebook that corresponds to the state having been matched with said true state.

15. (Previously Presented) The method of claim 13, wherein said signal segments of said codebook have variable lengths, each signal segment forming a trailing part of a signal frame, thereby enabling continuous transition from the expanded portion to a consecutive signal frame.

16. (Original) The method of claim 13, wherein time delays between said states in said codebook are incremental delays with a resolution of a fraction of a time between two samples.

17. (Original) The method of claim 14, wherein the states and the corresponding segments of said codebook are scaled in order to improve the matching with said true state.

18. (Original) The method of claim 14, wherein merging of said true state is performed with the matching state of said codebook.

19. (Previously Presented) The method of claim 14, wherein the second-listed producing step involves performing the corresponding operations with respect to a

heading part of the second signal frame being consecutive to the expanded portion.

20. (Previously Presented) The method of claim 26, wherein said first signal frame is either a sound signal frame resulting from a complete decoding operation of the first received frame, or an intermediate time-domain signal frame resulting from a partial decoding operation of the first received frame.

21. (Previously Presented) The method of claim 26, including the step of using an oscillator model, which oscillator model includes a codebook in which vectors of samples of a received digitized sound signal forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

22-25. (Canceled)

26. (Currently Amended) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that ~~looses~~ loses some packets, the method comprising steps of:

receiving a first received frame from the packet-switched network, wherein:

the first received frame is part of the received sound signal, and

the packet-switched network has packet loss;

producing a first signal frame corresponding to the first received frame, wherein:

the first signal frame is part of the sound signal, and

a second received frame is normally produced contiguously with the first received frame;

determining after beginning the first-listed producing step that the second received frame is currently unavailable for production; and

producing an expanded portion after the determining step, wherein:

the first signal frame and the expanded portion are contiguous parts of the sound signal,

the expanded portion corresponds to a different amount of the received sound signal than either the first or second received frames, and

the first signal frame and the expanded portion have different time lengths in the sound signal.

27. (Previously Presented) The method of claim 26, wherein the expanded portion is selected from the first signal frame based, at least in part, upon measures of periodicity.

28. (Previously Presented) The method of claim 26, wherein the determining step comprises a step of determining near the end of production of the first signal frame if the second received frame is currently unavailable for production.

29. (Previously Presented) The method of claim 26, further comprising steps of:

determining after beginning the second-listed producing step that the second received frame is still unavailable for production;

producing a second expanded portion after the immediately-preceding determining step, wherein the expanded portion and the second expanded portion are contiguous parts of the sound signal.

30. (Previously Presented) The method of claim 26, wherein:

a playback time of the expanded portion is variable, and

the playback time is selected based, at least in part, upon the sound signal.

31. (Previously Presented) The method of claim 26, wherein:

the first signal frame includes a plurality of sound samples, and

the expanded portion is determined with a time resolution finer than a sample rate of the plurality of sound samples.

32. (Previously Presented) The method of claim 26, further comprising a step of producing a second expanded portion based, at least in part, on some of the second received

frame, wherein the expanded portion and second expanded portion are contiguous parts of the sound signal.

33. (Previously Presented) The method of claim 26, further comprising a step of merging the expanded portion and a contiguous, subsequent, portion of the sound signal using a periodicity measure, whereby any audible discontinuities between the expanded portion and second expanded portion are reduced.

34. (Previously Presented) The method of claim 26, wherein the signal frame corresponds to a plurality of received frames.

35. (Previously Presented) The method of claim 26, further comprising a step of merging the expanded portion and a contiguous, subsequent, portion of the sound signal based, at least in part, on overlap-add, wherein a time shift of the first signal frame and expanded portion is optimized based, at least in part, on correlation.

36. (Previously Presented) The method of claim 26, further comprising steps of

measuring overload of a jitter buffer;

discarding some of the second received frame based, at least in part, on the overload; and

merging a preceding and a subsequent portions of the sound signal after the discarding step.

37. (Previously Presented) The method of claim 36, further comprising steps of:

determining if a signal fitting criteria between the preceding and subsequent portions is fulfilled; and

performing the discarding step only with the immediately-preceding determining step is fulfilled.

38. (Previously Presented) The method of claim 36, wherein a length of the some of the second received frame is based, at least in part, on the sound signal.

39. (Previously Presented) The method of claim 36, wherein the some of the second received frame comprises a plurality of sub-portions that are sequentially discarded.

40. (Previously Presented) The method of claim 36, wherein:

the merging step is based, at least in part, on overlap-add, and

any time-shift of the preceding and subsequent portions is optimized based, at least in part, on a measure of periodicity.

41-42. (Cancelled)

43. (Currently Amended) A method for manipulating a received sound signal to product; a sound signal, wherein the received sound signal is received from a packet-switched network that ~~looses~~ loses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein the first signal frame is part of the sound signal;

determining after beginning the first-listed producing step that a second received frame currently unavailable for production due to latency;

producing a first expanded portion after the first-listed determining step, wherein:

the first expanded portion and the first signal frame are contiguous parts of the sound signal,

the first signal frame and the second signal frame would be contiguous parts of the sound signal in situations where the second received frame is available for production, and

the first expanded portion has a different size than either the first or second received frames;

receiving a third received frame that is part of the received sound signal;

producing a third signal frame corresponding to the third received frame, wherein the third signal frame is part of the sound signal;

determining after beginning the second-listed producing step that a fourth received frame currently unavailable for production due to packet loss; and

producing a second expanded portion after the second-listed determining step, wherein:

the second expanded portion and the third signal frame are contiguous parts of the sound signal,

the third signal frame and the fourth signal frame would be contiguous parts of the sound signal in situations where the fourth received frame is available for production,

the second expanded portion has a different size than either the third or fourth received frames, and

the first and third signal frames have a frame size that is different from a size of the first expanded portion.

44. (Currently Amended) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that ~~looses~~ loses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein:

the first signal frame is part of the sound signal, and

a second received frame is produced contiguously with the first received frame when the second received frame is available;

determining after beginning the first-listed producing step that the second received frame is currently unavailable for production due to packet loss; and

producing an expanded portion after the determining step, wherein:

the first signal frame and the expanded portion are contiguous parts of the sound signal, and

the expanded portion has a size that is different than a frame size of the first signal frame.

45. (Previously Presented) The method of claim 26, further comprising a step of producing a second signal frame corresponding to the second received frame, wherein the expanded portion and the second signal frame are contiguous parts of the sound signal.

46. (Previously Presented) The method of claim 26, further comprising a step of time shifting the expanded portion relative to the first signal frame whereby periodicity of the sound signal is improved across a boundary between the first signal frame and the expanded portion.

47. (Previously Presented) The method of claim 26, further comprising a step of producing a second signal frame corresponding to the second received frame, wherein the first and second signal frames have a first frame size that is different from a size of the expanded portion.

48. (Previously Presented) The method of claim 43, wherein the second expanded portion has a second size different from the frame size.

49. (Previously Presented) A method for manipulating a sequence of digitized sound signal frames of a sound signal, the sound signal frames being decoded from packet data received from a packet switched network, the method including:

decoding a first sound signal frame from the packet data received from the packet switched network;

determining that, due to the occurrence of packet loss, the packet data is not available for decoding a second sound signal frame which should be contiguous with the first sound signal frame; and

producing an expanded frame portion to be contiguous with the first sound signal frame, wherein the expanded frame portion represents a part of the sound signal that is different from the part represented by the first signal frame, and wherein the time length of the expanded frame portion in the sound signal is different from the time length of first sound signal frame, and wherein a following frame decoded from the packet data received from the packet switched network is provided to be contiguous with the expanded frame portion.

50. (Previously Presented) The method as claimed in claim 49, wherein the time length of the expanded frame portion is chosen such that it provides a smooth transition to said following frame.

51. (Currently Amended) The method as claimed in claim 49, wherein the time length of the expanded frame portion frame is chosen based upon ~~the~~ a requirement to ~~fulfil~~ fulfill a signal fitting criteria with respect to ~~the~~ signal characteristics of the sound signal.

52. (Currently Amended) The method as claimed in claim 49, wherein ~~the~~ a resolution of the time length of the expanded frame portion is a fraction of the time between two samples of said sound signal, thereby providing an improved signal fitting quality for a smooth transition to said following frame.

53. (Previously Presented) The method as claimed in claim 49, wherein the determining step includes monitoring of a jitter buffer which stores received data packets to be decoded into signal frames.

54. (Previously Presented) The method as claimed in claim 49, including matching a true state of a trailing part of the first sound signal frame with an entry state of a codebook, the codebook having entries consisting of different states formed by vectors of samples and storing a corresponding signal segment for each entry state, wherein the step of producing an expanded frame portion includes reading out a signal segment from said codebook that corresponds to the entry state that have been matched with said true state.

55. (Previously Presented) The method as claimed in claim 54, wherein different signal segments of said codebook have different time lengths, each signal segment forming a trailing part of a signal frame, thereby enabling continuous transition from the time expanded signal frame to a consecutive signal frame.

56. (Previously Presented) The method as claim 54, wherein time delays between states of said entries in said codebook are incremental delays with a resolution of a fraction of a time between two samples.

57. (Previously Presented) The method as claimed in claims 54, wherein the states of said codebook are scaled in order to improve the matching with said true state, and wherein a trailing

part of the signal segment read out from the codebook is scaled to provide a smooth transition to said following frame.

58. (Cancelled)

59. (Currently Amended) ~~An apparatus~~ A system for manipulating a sequence of digitized sound signal frames of a sound signal, the sound signal frames being decoded from packet data received from a packet switched network, the ~~apparatus~~ system including:

a memory element for storing a computer program and vectors of samples of the received sound signal together with corresponding signal segments; and

a processor unit for executing a computer program causing the ~~apparatus~~ system to:

decode a first sound signal frame from the packet data received from the packet switched network;

determine that, due to the occurrence of packet loss, the packet data is not available for decoding a second sound signal frame which should be contiguous with the first sound signal frame; and

produce an expanded frame portion to be contiguous with the first sound signal frame, wherein the expanded frame portion represents a part of the sound signal that is different from the part represented by the first signal frame, and wherein the time length of the expanded frame portion in the sound signal is different from the time length of first sound signal frame, and wherein a following frame decoded from the packet data received from the packet switched network is provided to be contiguous with the expanded frame portion.

60. (New) A system for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that loses some packets, the system comprising:

a timing recovery and lost frame substitution for

producing a first signal frame corresponding to a first received frame from the packet-switched network, wherein

the first received frame is part of the received sound signal,

the packet-switched network has packet loss,

the first signal frame is part of the sound signal, and

a second received frame is normally produced contiguously with the first received frame;

determining after beginning producing the first signal frame that the second received frame is currently unavailable for production, and

producing an expanded portion after determining that the second received frame is currently unavailable for production, wherein

the first signal frame and the expanded portion are contiguous parts of the sound signal,

the expanded portion corresponds to a different amount of the received sound signal than either the first or second received frames, and

the first signal frame and the expanded portion have different time lengths in the sound signal.

61. (New) The system of claim 60, wherein use is made of an oscillator model for extracting signal segments from the first signal frame, the oscillator model including a codebook in which vectors of samples forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

62. (New) The system of claim 61, wherein the timing recovery and lost frame substitution produces the expanded portion by matching a true state of a trailing part of the first signal frame with said states in said codebook, and reading out a signal segment from said codebook that corresponds to the state having been matched with said true state.

63. (New) The system of claim 61, wherein said signal segments of said codebook have variable lengths, each signal segment forming a trailing part of a signal frame, thereby enabling continuous transition from the expanded portion to a consecutive signal frame.

64. (New) The system of claim 61, wherein time delays between said states in said codebook are incremental delays with a resolution of a fraction of a time between two samples.

65. (New) The system of claim 62, wherein the states and the corresponding segments of said codebook are scaled in order to improve the matching with said true state.

66. (New) The system of claim 62, wherein merging of said true state is performed with the matching state of said codebook.

67. (New) The system of claim 62, wherein the timing recovery and lost frame substitution produces the expanded portion by performing the corresponding operations with respect to a heading part of the second signal frame being consecutive to the expanded portion.

68. (New) The system of claim 60, wherein said first signal frame is either a sound signal frame resulting from a complete decoding operation of the first received frame, or an intermediate time-domain signal frame resulting from a partial decoding operation of the first received frame.